3/1978

Concise Papers

Also see Appln Pg 3

Hybrid D-PCM, A Combination of PCM and DPCM

M. C. W. VAN BUUL

Abstract—After a short review of the basic principles of PCM and DPCM coding it is shown that combining the two methods in one coding system results in a combination of some favorable properties. The hybrid D-PCM system combines the bit-rate reduction of DPCM with a low error sensitivity. In another approach the hybrid D-PCM system is also derived from a usual DPCM system with a leaking integrator. Two different applications of the hybrid D-PCM system illustrate the possibilities and the performance of the hybrid D-PCM system.

I. INTRODUCTION

The most obvious method for the digital encoding of television signals is linear pulse code modulation or PCM. However for the transmission or storage of a digitized television signal it is desirable to reduce the number of bits of the encoded signal as much as possible without losing too much relevant information. A rather efficient method, which can be implemented quite simply, is differential PCM or DPCM.

In a PCM system the incoming signal is sampled and the amplitude of each sample is measured with a fixed scale, as illustrated in fig. 1. The distinct levels on the scale can have a linear or a nonlinear distribution and they are numbered in order, starting with a fixed zero-level. The amplitude of each sample is digitized by rounding off its amplitude to the nearest distinct scale level (quantizing) and assigning the appertaining number to the sample.

These numbers can be processed or transmitted and each number is easily reconverted in the PCM decoder to a sample amplitude that corresponds to the original quantized level on the measuring scale. When a transmission error in the case of a digitized television signal results in a wrong number at the decoder, the decoded picture shows only one wrong picture element on the display, which in general is not very disturbing.

In a DPCM system the amplitude of the incoming samples is measured with a sliding scale, as illustrated in fig. 2. Here the zero level of the scale is put at the quantized amplitude of the previous sample and the distinct levels on the scale are again numbered in order. Each sample amplitude is then measured

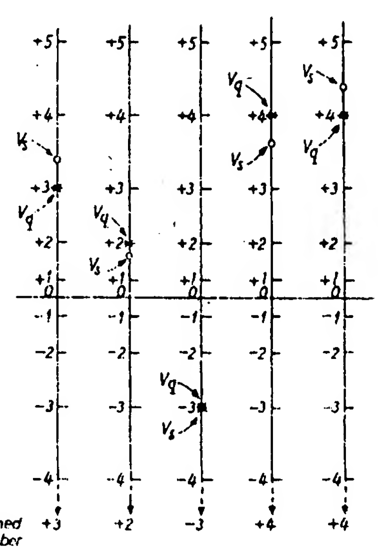

Fig. 1. Principle of PCM coding; quantizing and coding with a fixed measuring scale.
○ = sampled signal amplitude V_s.
• = quantized sample amplitude V_q.

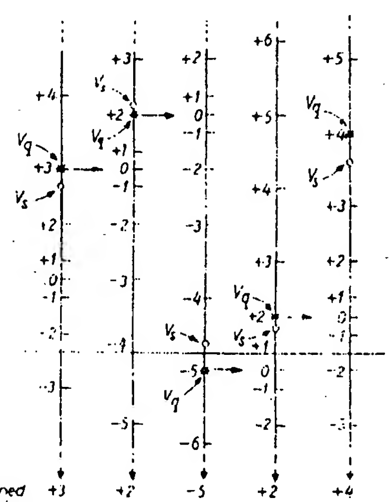

Fig. 2. Principle of DPCM coding; quantizing and coding with a sliding measuring scale. The zero of the scale is determined by the quantized amplitude of the previous sample.
○ sampled signal amplitude V_s.
• quantized sample amplitude V_q.

Paper approved by the Editor for Communication Electronics of the IEEE Communications Society for publication after partial presentation at the 1976 Picture Coding Symposium, Asilomar, CA. Manuscript received February 18, 1977; revised July 19, 1977.

The author is with Philips Research Laboratories, Eindhoven, The Netherlands.

0090-6778/78/0300-0362$00.75 © 1978 IEEE

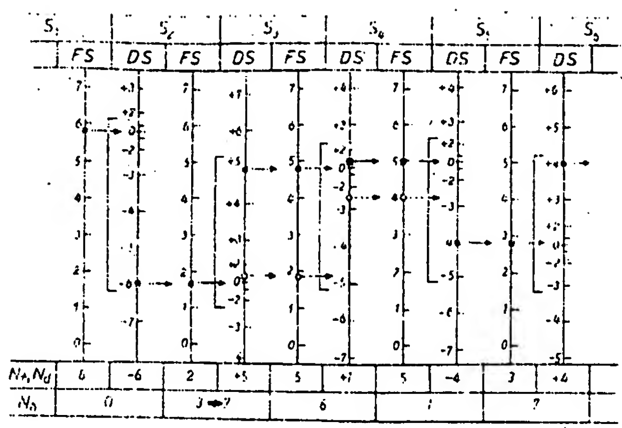

Fig. 4. Error correction of hybrid D-PCM coding; the error is made at the transmission of S_3.
FS, DS, N_f, N_d, N_h = fig. 3.
• = actual sample amplitude (partly erroneous).
○ = correct sample amplitude (when the actual sample amplitude is erroneous).

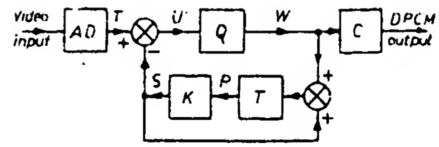

Fig. 5. Normal DPCM encoder with a leaking integrator (digital implementation). The input signal V_U of the quantizer Q is:
$V_U = V_T - V_S = (V_T - V_R) + (1-k)V_R$.
AD = analog-to-digital converter (PCM).
Q = quantizer.
C = code converter.
T = delay unit.
K = multiplier.

difference signal, which again is added to the amplitude of sample S_2, resulting in the amplitude of sample S_3. In this way all sample amplitudes are reconstructed from the incoming sequence of numbers.

From the foregoing it appears that the recovery of the difference number N_d in the decoder does not depend on the actual value of the PCM number N_f, as long as in the encoder and the decoder the same numbers N_f occur (encoder: $N_h = N_f + N_d \Rightarrow$ decoder: $N_d = N_h - N_f$). This means that the accuracy of the signal reconstruction in the decoder is independent of the kind of quantization of the previous sample amplitude that is used to obtain N_f. So for the reconstruction we can tolerate indeed a very coarse fixed scale FS, as the one used in the example of fig. 3. The choice of the scale FS only influences the error correction (section 2.3) and the limitation of N_d (section 2.4).

2.3 The error correction

Fig. 4 shows the response of the hybrid D-PCM decoder of fig. 3 to a transmission error. It is assumed that the second number $N_{h2} = 3$ is received as $N_{h2} = 7$, which corresponds to an inversion of the most significant bit. In exactly the same way as in fig. 3 the sequence of the reconstructed sample amplitudes can be obtained from the incoming sequence of numbers N_h. For easy comparison the correct sample amplitudes are indicated in fig. 4 with small circles and dashed arrows. In this example the bit error is corrected completely after only two sample periods, which is quite fast compared with a conventional DPCM system.

In general, large errors are reduced quite fast, but small errors, which fortunately are much less disturbing, have a much slower decay. With a constant input signal even a permanent error at the decoder may remain after a transmission error. This can be understood by recalling that the PCM information in the transmitted hybrid D-PCM numbers N_h is coarsely quantized. When the output signal of the decoder is so close to the input signal of the encoder that both give rise to the same number N_f from the measurement with the coarse PCM scale FS, then no further error correction is performed until the input signal of the encoder varies so much that the numbers N_f in the encoder and in the decoder show a difference again.

A similar, generally concealed effect occurs in a digitally implemented usual DPCM system with leaking integrator (compare section 3) caused by rounding in the multiplier (K in fig. 5) or the succeeding subtractor.

It can be noted here that the error correcting property of the hybrid D-PCM decoder justifies the initial assumption in section 2.2, that the previous sample amplitude was decoded correctly. Any difference between the sample amplitude in the encoder and its reconstructed amplitude in the decoder will be corrected as if it were a transmission error.

2.4. The effects of limiting N_d

As indicated before, the desired limitation of the hybrid D-PCM number N_h from 0 to +7 is obtained by a suitable limitation of the DPCM number N_d. This means that the range of the possible levels on the difference scale DS is determined by the amplitude of the previous sample. If this previous sample has a low amplitude, the resulting PCM number N_f has a small value, so that only a small negative transition is possible but a much larger positive one. For instance sample $S3$ in fig. 3 gives rise to a PCM number $N_f = 2$, which allows the DPCM number N_d to acquire a value from -2 (small negative transition) up to $+5$ (large positive transition). Conversely, when the previous sample has a high amplitude, then only a small positive and a much larger negative transition are possible. So the hybrid D-PCM system provides an "automatic" adaptation of the range of the difference signal to the amplitude of the input signal. This is illustrated very clearly in figs. 3 and 4, where at each sample the allowed range for the coding of the next sample is indicated. The effect is similar to the result obtained with a switched quantizer [2], [3].

From the figs. 3 and 4 it also appears that a severe overload can occur when the input signal already has a rather low or a rather high amplitude (e.g., $N_f = 0$; 1; 6 or 7) and it shows a small transient to an even more extreme value. This effect can be improved by a better adaptation of the quantization characteristics of the scales FS and DS to each other, as in the first application described in section 4. Another possibility is to expand the scale FS to such an extent that the extreme numbers of FS (e.g., $N_f = 0$; 1; 6 and 7) are outside the amplitude range of the input signal, as in the second application in section 4. In the latter case some part of the adaptation of the range of the difference signal to the amplitude of the input signal is lost also, but still a considerable improvement with respect to conventional DPCM systems is preserved.

3. THE LEAKING INTEGRATOR APPROACH

A common method for reducing the error propagation of a conventional DPCM system is the use of a leaking integrator, both in the decoder and in the encoder. Fig. 5 shows a block diagram of a digitally implemented DPCM encoder in which the leak in the integrator can be obtained by multiplying the output V_R of the delay unit T by a constant factor k. With $k = 1$ the integrator has no leak (ideal integrator) and the input signal of the quantizer Q is

$$V_U = V_T - V_S = V_T - V_R$$

When $k < 1$, V_U can be expressed as

$$V_U = V_T - V_S = V_T - kV_R = (V_T - V_R) + (1-k)V_R.$$

The input signal V_U of the quantizer Q now consists of the original difference signal $(V_T - V_R)$ plus some information about the real amplitude V_R of the previous sample. This additional information about the real amplitude of a sample enables a decoder to correct for a transmission error, and the correction will be performed faster as more of this information is available. However, when we increase this information by decreasing the multiplication factor k, then the original operating point of the quantizer Q shifts from the center of the transfer characteristic to a point where small differences are more coarsely quantized. Fig. 6 shows a rather extreme example of this shift. Because of the coarser quantization the contouring effect and granular noise of the DPCM encoder increase considerably with respect to the normal operating point. So in practice we have to compromise between these effects and a fast error correction.

The proposed hybrid D-PCM system as depicted in fig. 7 avoids this compromise by adding the information about the real sample amplitude in another way. Instead of passing this additional information through the quantizer Q, as with the leaking integrator of fig. 5, the information is passed through a separate branch and it is added to the difference information only after the quantizer Q and code converter C, as shown in fig. 7. This means that an encoder can be used with an ideal integrator in the feedback loop, as shown within the dashed lines. Because of this ideal integrator in the encoder the operating point of the quantizer Q is no longer shifted, and via the other branch much more information about the real sample amplitude can be added without introducing any additional contouring or granular noise.

Because the number of bits in the transmission channel has been reduced in a DPCM system, the signal at the output of the code converter C usually has fewer bits per sample than the signal at the output of the delay unit T. Therefore in the hybrid D-PCM system an additional quantizer Q' and coder converter C' are used to adapt the number of bits before the PCM part of the hybrid D-PCM signal is added to the DPCM part.

In the hybrid D-PCM decoder (fig. 8) the additional information about the real amplitude of the previous sample, which has been added to the difference signal in the encoder, is subtracted from the incoming hybrid D-PCM signal and the recovered normal DPCM signal is decoded in the usual way.

In the block diagram of the hybrid D-PCM decoder the integrator is seen to be embodied in a feedback loop. This feedback loop causes the integrator to follow the input signal of the decoder rather closely, so that the effect of a single transmission error is rapidly reduced. The actual decay of an error depends both on the choice of the transfer characteristics of the quantizers Q and Q' and on the actual input signal.

It can easily be seen that the block diagram of fig. 7 indeed represents an implementation of the procedure for the hybrid D-PCM encoding system as described in section 2.1. The normal DPCM encoder within the dashed lines performs the difference measurement; the additional quantizer Q' and code converter C' perform the coarse PCM encoding of the amplitude of the previous sample. The two numbers resulting from these two measurements are added to obtain the hybrid D-PCM number (compare with fig. 3: $N_h = N_d + N_f$).

In the block diagram of the hybrid D-PCM decoder the PCM number N_f belonging to the previous sample is indeed subtracted from the incoming hybrid D-PCM number N_h to give the DPCM number N_d, which is decoded in the usual way.

As indicated before, the transfer characteristics of the quantizers Q and Q' determine the error correction and overload performance of the hybrid D-PCM system. Fig. 9 shows the

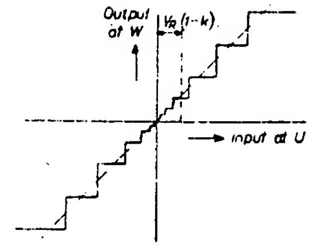

Fig. 6. Transfer characteristic of the quantizer Q. The leak of the integrator causes a shift of the operating point from the center to the point $(1-k)V_R$.

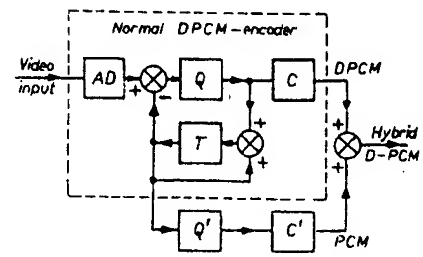

Fig. 7. Block diagram of a hybrid D-PCM encoder. An ideal integrator is used in the feedback loop, so the operating point of Q is not shifted.

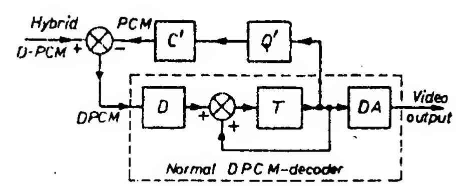

Fig. 8. Block diagram of a hybrid D-PCM decoder. The "normal DPCM decoder" is embodied in a feedback loop, so that transmission errors are rapidly corrected.

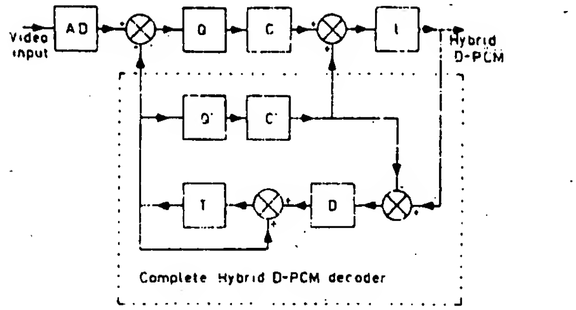

Fig. 9. Block diagram of a hybrid D-PCM encoder, containing a complete decoder and a limiter L.

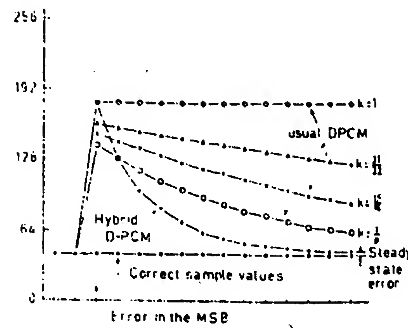

Fig. 11. Error correction of the hybrid D-PCM system with the characteristics of fig. 10, compared with usual DPCM systems.

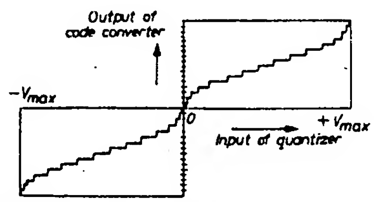

Fig. 10. Well-suited transfer characteristic for the quantizer and code converter of a hybrid D-PCM system.

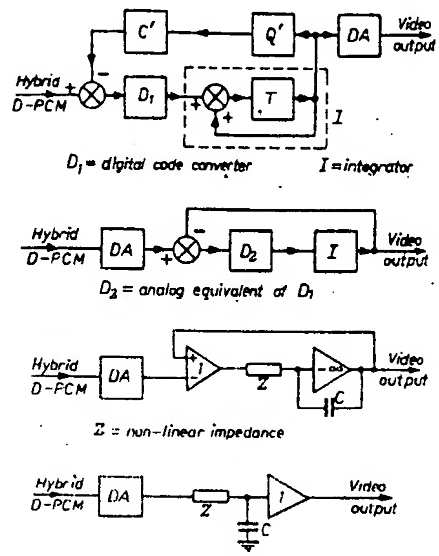

Fig. 12. Derivation of a hybrid D-PCM decoder with mainly analog signal processing from a fully digital decoder.

block diagram of an implementation, that performs the desired limiting of N_h and that allows a complete freedom in the choice of the characteristics of Q and Q'. This is obtained by incorporating a complete decoder (compare fig. 8) in the feedback loop, so that an accumulation of the effects of limiter L is prevented.

4. APPLICATIONS

Fig. 10 shows a type of transfer characteristic for the quantizer and code converter, that is very well suited for use in the hybrid D-PCM system. This type of quantizing characteristic has been proposed before by Bostelmann [4]. The advantages of his proposal are that no slope overload occurs and that in the transmission one bit is saved as compared with usual DPCM systems. If the transfer characteristics of the additional quantizer Q' and code converter C' in the hybrid D-PCM system (see fig. 7) are chosen to be equal to the positive part of the characteristics of the differential quantizer Q and code converter C, then exactly the same advantages are obtained as with Bostelmann's original proposal. This means also that the slope overload as described in section 2.4 is completely eliminated. Additionally, however, a fast error correction is obtained, which is shown in fig. 11. For an easy comparison, fig. 11 also shows the error responses of a usual DPCM system with an integrator with various leak factors k (see section 3).

From this example it will be clear that, with a suitable choice of the quantizing characteristics, the error performance of the hybrid D-PCM system is far superior to that of usual DPCM systems. Moreover, in contrast with systems with a leaking integrator, the hybrid D-PCM system does not increase the granular noise and contouring in flat areas of high or low signal values. In the example of fig. 11 it appears also that with a constant input signal a small steady state error may remain (compare with section 2.3). Computer simulation of the system has shown, however, that with a varying input signal the error is usually completely corrected within a few samples.

A second interesting application of the hybrid D-PCM system is a very simple encoder and decoder with mainly analog signal processing. Fig. 12 shows how an analog decoder can be derived from a fully digital decoder. First the digital-to-analog converter is moved from the output of the decoder to the front end, and a linear quantizing characteristic is chosen for the PCM feedback part. Next, the digital integrator is replaced by an analog one and the digital code converter D_1 is replaced by its analog equivalent D_2. The third diagram of fig. 12 shows how in principle the functions of the second diagram can be implemented. The integrator consists of an operational amplifier with a feedback capacitor C.

Because the input of the operational amplifier is at virtual ground, the voltage across the nonlinear impedance Z is the voltage difference between the output of the DA converter and the video output. The current through Z, caused by this voltage difference, is integrated in the capacitor C. Finally the fourth diagram shows an equivalent implementation. Again the voltage difference between the DA converter and the video output signal is put across the nonlinear impedance Z and the current through Z is integrated in the capacitor C.

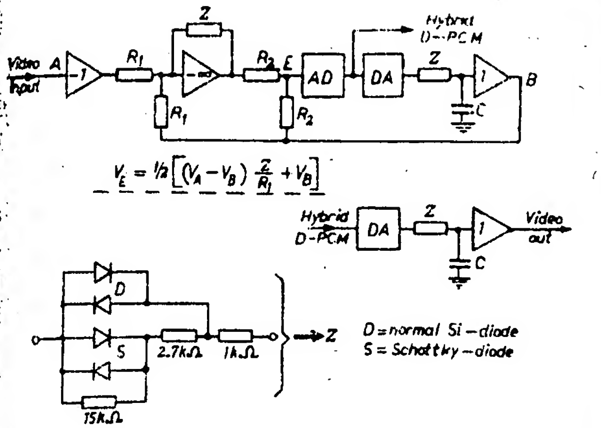

Fig. 13. Hybrid D-PCM encoder and decoder based on the decoder of fig. 12 and the encoder of fig. 9.

Fig. 13 shows how a hybrid D-PCM encoder can be made by embodying the analog decoder of fig. 12 in a feedback loop in a similar way as in fig. 9. The formula shows that the signal V_E at the input of the AD converter is indeed composed of the output signal V_B of the decoder (the amplitude of the previous sample) and the nonlinearly amplified difference signal ($V_A - V_B$). For proper operation the AD and DA converter together ought to have a gain of two.

From these two applications it will be clear that the hybrid D-PCM system can be used for many types of predictive coders. Also for coders that use both horizontal and vertical prediction, the hybrid D-PCM system is able to reduce the visibility of transmission errors considerably.

5. EXPERIMENTS

As a test for the hybrid D-PCM system the encoder and decoder of fig. 13 have been built. The actual implementation of the AD/DA converter combination has been inspired by the converter of [5] (fig. 7 in [5]) and for slowly varying input signals the operation of the circuit of fig. 13 shows a remarkable similarity to the corresponding part of the circuit in [5]. Actually fig. 5 of [5] holds identically for the hybrid D-PCM system of fig. 13. Only for transients in the input signal the nonlinearity of the impedances Z is important. The resistances in Z (lower part of fig. 13) determine the ratio of the small, medium, and large step sizes of the difference signal (scale DS in section 2) and the threshold voltages of the diodes determine the transitions between the different step sizes. The value of capacitor C depends on the sampling frequency of the AD/DA converter and it determines together with R_1 (actually 1 kΩ) the granular noise and transient response of the system.

In fig. 14 a picture is shown that has been encoded and decoded by the hybrid D-PCM system of fig. 12. The sample frequency chosen is 256 times the horizontal line frequency, as is usual with videophone systems, and 4 bits per sample are used.

The signal amplitude in the encoder has been adjusted such that only the eight levels in the center of the 4-bit AD converter are used for the coding of the previous sample amplitude V_B. This avoids the slope overload for small transients at extreme signal amplitudes, as described in section 2.4. Due to

Fig. 14. Result of the encoding and decoding of a videophone picture with the hybrid D-PCM system of fig. 13.

Fig. 15. See fig. 14. The performance of the error correction is shown by introducing an inversion of the most significant bit at the same position in each line.

the automatic adaptation of the range of the difference signal to the level of the input signal, large and steep transients show no annoying slope overload. The effective range of the difference scale is about from −12 to +12.

Fig. 15 shows the effect of a transmission error on the hybrid D-PCM system. At the same position in each line an inversion of the most significant bit has been introduced. It is obvious that these errors are rapidly corrected. The performance of the error correction is comparable with a usual DPCM system with an integrator time constant of about four sample periods. With the hybrid D-PCM system, however, no additional contouring and granular noise are introduced at all (see section 3)

6. CONCLUSION

The hybrid D-PCM system presented in this paper is a simple and very effective means of reducing the error sensitiv-

ity of predictive coders for television signals. This is achieved by a suitable incorporation of PCM information in the conventional difference code without increasing the bit rate. Moreover the automatic adaptation of the allowed range of the difference signal to the actual level of the input signal considerably reduces the slope overload of large steep transitions without increasing granular noise and contouring effects.

REFERENCES

[1] D. J. Connor, Techniques for reducing the visibility of transmission errors in digitally encoded video signals. *IEEE Trans. Commun.* COM-21 (1973), no. 6, pp. 695-706.

[2] H. G. Musmann, Codierung von Videosignalen. *Nachrichtentechn. Z.* 24 (1971), H2, pp. 114-116.

[3] H. H. Bauch *et al.*, Picture coding. *IEEE Trans. Commun.* COM-22 (1974), no. 9, pp. 1158-67.

[4] G. Bostelmann, A simple high quality DPCM-codec for video telephony using 8 Mbit per second. *Nachrichtentechn. Z.* 27 (1974), H. 3, pp. 115-117.

[5] J. C. Candy, A use of limit cycle oscillations to obtain robust analog-to-digital converters. *IEEE Trans. Commun.* COM 22 (1974), no. 3, pp. 298-305.

Autonomous Line Scanning for SPC Telephone Switching Systems

A. K. SHRIVASTAVA, SENIOR MEMBER, IEEE, P. RAMANADHAN, A. V. SHAH, MEMBER, IEEE, AND H. S. JAMADAGNI

Abstract—An Autonomous Line Scanning Unit (ALSU) for completely autonomous detection of call originations in the SPC Telephone Switching System is described. Through its own memories, ALSU maintains an up-to-date record of subscribers' statuses, detects call originations, performs 'hit timing check' and informs the Switching System of the identity of calling subscribers. The ALSU needs minimum interaction with the Central Processor, resulting in increased call handling capacity.

Paper approved by the Editor for Communication Switching of the IEEE Communications Society for publication without oral presentation. Manuscript received February 4, 1977; revised October 3, 1977. This work was the result of research carried out jointly at Indian Telephone Industries Ltd., Bangalore, India, and at the Centre for Electronics Design Technology set up at Indian Institute of Science, Bangalore, in collaboration with the Swiss Federal Institute of Technology, Zurich.

A. K. Shrivastava and P. Ramanadhan are with the Switching Research and Development Division, Indian Telephone Industries Ltd., Bangalore, India.

A. V. Shah and H. S. Jamadagni are with the Centre for Electronics Design Technology, Department of Electrical Communication Engineering, Indian Institute of Science, Bangalore, India.

I. INTRODUCTION

The Telephone Network of India, at the present stage, consists of electromechanical systems only. About 80% of subscribers are being served by automatic exchanges of Strowger or Crossbar types. Electronic Switching Systems are in advanced stage of development. An experimental 100 line Stored Program Controlled (SPC) Electronic Telephone Exchange has been tried out successfully in the Telecommunication Research Centre of Indian P & T, and Indian Telephone Industries is manufacturing a 1000 line SPC Electronic Telephone Exchange for field trials. Telephone traffic in India has peculiar characteristics. The calling rate and average both way traffic in some metropolitan areas may go as high as 16 Busy Hour Calls and 8.64 CCS (0.24 Erlang) per line. Telephone exchanges here are often subjected to very heavy traffic loads. Methods of increasing the call handling capacity of telephone switching systems, therefore, assume great importance in India. Enhancing call handling capacity of the system by changes in the architecture of its Central Processor (CP) or by replacing the CP with a faster one would generally be more expensive than off-loading the CP of certain functions by using a front-end, special purpose processor. One such front-end processor unit is described in this paper.

In a telephone exchange there are some functions of a routine nature which do not require flexibility. Such functions, if done under the control of stored program, would add a considerable load to CP without really needing its intelligence [1, 2]. Detection of originating calls is one such function which can be done independently by an Autonomous Line Scanning Unit (ALSU) with its own wired logic control. Some form of *Autonomous Scanning* has been reported in the following systems:

(a) No. 2 ESS (Bell System, USA) [3],
(b) No. 4 ESS (Bell System, USA) [4],
(c) AKE 132, Transit Exchange System (LM Ericsson, Sweden) [5],
(d) SP-1, 4 Wire ESS (Northern Telecom Limited, Canada) [6],
(e) PRX System (Philips, Holland) [7],
(f) No. 1 EAX System (GTE Automatic Electric Inc., USA) [8],
(g) KDX-0 System (Kokusai Denshin Denwo Co. Ltd., and Nippon Electric Co. Ltd., Japan) [9],
(h) System 250 (Plessey, England) [10].

These systems do not handle the complete process of originating call detection autonomously and substantial processor interaction is necessary in most of the cases. Characteristic features of ALSU that distinguish it from the systems mentioned above are: (i) Completely autonomous detection of call originations, (ii) 'Hit Timing Check' on detected originations without CP interaction, (iii) Listing of calling subscribers' identities within the unit for transfer to CP under interrupt, (iv) No need for CP to maintain a record of subscribers' statuses, since these can be obtained from the unit.

Concepts of the *SPC Telephone Switching System* (Fig. 1) [11-13] may be recapitulated here for the sake of clarity. In this system, calls are processed by a CP which is a Stored Program